

Amendments to the Claims

1. (Previously presented) An audio signal processing apparatus in which a waveform of a digital audio signal to be replayed is processed, the apparatus comprising:

frequency bandwidth expanding means for expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

low-pass filtering means for performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting means for detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

difference data calculating means for calculating data of a difference between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting means for weighting the data of the difference depending on the interval of time between the two adjacent waveform peaks; and

producing means for producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference.

2-3. (Canceled)

4. (Previously presented) The audio signal processing apparatus of claim 1, wherein the past data of the low-pass-filtered digital audio signal used by the difference data calculating means are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency.

5. (Previously presented) The audio signal processing apparatus of claim 4, wherein the weighting means is configured to weight the difference data depending on both the interval of time and the polarities of the gradients.

6. (Previously presented) The audio signal processing apparatus of claim 4, wherein the producing means is configured to add the weighted difference data to the low-pass-filtered digital audio signal.

7. (Previously presented) A method of processing a waveform of a digital audio signal to be replayed, comprising steps of:

expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

calculating data of a difference between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting the data of the difference depending on the interval of time between the two adjacent waveform peaks; and

producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference.

8. (Canceled)

9. (Previously presented) The processing method of claim 7, wherein the past data of the low-pass-filtered digital audio signal are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency.

10. (Previously presented) The processing method of claim 9, wherein the weighting step weights the difference data depending on both the interval of time and the polarities of the gradients.

11. (Currently amended) A computer-readable program stored in a computer readable medium and executed ~~used~~ by a computer for processing a waveform of an inputted digital audio signal to be replayed, the program allowing the computer to functionally realize steps of:

- expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

- performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

- detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

- calculating data of a difference between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

- weighting the data of the difference depending on the interval of time between the two adjacent waveform peaks; and

- producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference.

12. (Previously presented) The audio signal processing apparatus of claim 1, wherein the producing means adds the weighted difference data to the low-pass-filtered digital audio signal.
13. (Previously presented) The processing method of claim 9, wherein the producing step adds the weighted difference data to the low-pass-filtered digital audio signal.
14. (Previously presented) The computer-readable program of claim 11, wherein the past data of the low-pass-filtered digital audio signal are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency.
15. (Currently amended) The computer-readable program of claim ~~13~~ 14, wherein the weighting step weights the difference data depending on both the interval of time and the polarities of the gradients.
16. (Currently amended) The computer-readable program of claim ~~13~~ 14, wherein the producing step adds the weighted difference data to the low-pass-filtered digital audio signal.